SPEECH DATA RECEIVER WITH DETECTION OF CHANNEL CODING RATE

DESCRIPTION

Background Art

The following acronyms may be used throughout this description. They are listed in TABLE 1 below for ease of reference.

TABLE 1

ACRONYM	**Definition**
ACS	Active Codec Set
AFS	AMR Full rate Speech service
AHS	AMR Half rate Speech service
AMR	Adaptive Multi Rate speech service
ASIC	Application Specific Integrated Circuit
BER	Bit Error Rate
BSS	Base Station Subsystem
BTS	Base Transceiver Station
CDMA	Code Division Multiple Access
CHD	Channel Decoder
CHE	Channel Encoder
C/I	Carrier-to-Interference ratio (used to measure link quality)
CMI	Codec Mode Indication (speech rate used on attached link)
CMC	Codec Mode Command (speech rate commanded to be used by an MS on its uplink)
CMR	Codec Mode Request (speech rate requested by an MS to be used on its receiving link)
CRC	Cyclic Redundancy Check
dB	decibels
DL	Downlink
DSP	Digital Signal Processor
DTX	Discontinuous Transmission
EFR	Enhanced Full Rate speech codec for GSM

ACRONYM	Definition
EVRC	Enhanced Variable-Rate Codec, used in IS-95 CDMA
FEC	Forward Error Correction
FACCH	Fast Associated Control Channel
FER	Frame Erasure Rate
FPGA	Field Programmable Gate Array
FR	Full Rate speech codec for GSM
GSM	Global System for Mobile communications, common digital cellular standard
HR	Half Rate speech codec for GSM
KBPS	Kilo Bits Per Second
MS	Mobile Station, e.g. a cellular phone
RATSCCH	Robust AMR Traffic Synchronized Control Channel
RBER	Residual Bit Error Rate
RF	Radio Frequency
RXQUAL	Received Signal Quality
SID	Silence Descriptor
SID_UPDATE	AMR Frame Used to Convey Comfort Noise Characteristics During DTX
SNR	Signal to Noise Ratio
SPD	Speech Decoder
SPE	Speech Encoder
TDMA	Time Division Multiple Access
TRAU	Transcoding and Rate Adapting Unit
UL	Uplink

Currently, the primary usage of digital cellular systems is for the transmission of voice. The limited available spectrum (bandwidth) of such systems requires that speech be encoded using a minimal number of bits in order to reduce the redundancy of the source data. Potentially poor channel conditions typical in cellular systems, e.g. low SNR and fading, necessitate the use of a channel coding scheme to add redundancy back in an efficient manner. Typically, the channel coding consists of a

forward error correction scheme (block or convolutional code) and an error detection scheme, e.g. CRC.

Within the context of the GSM digital cellular standard, several speech codecs are standardized and in use. The original GSM speech codec is commonly referred to as the Full-Rate (FR) speech codec and encodes speech at a rate of 13 kbps. The next generation of codecs took divergent paths. The Half-Rate (HR) codec allowed for a doubling of system capacity but at the expense of voice quality. The Enhanced Full-Rate (EFR) kept the speech rate approximately the same (12.2 kbps), but improved algorithms and increased DSP processing power provided significantly higher voice quality. This codec has been well received and is currently used in most GSM systems. All of these voice services use convolutional codes for error correction and some form of CRC for error detection.

In 1997, the process of standardizing a new GSM speech service was begun in order to take advantage of speech coding advances. A set of requirements was established that included both quality and capacity increases over previous GSM codecs. The improved quality requirements primarily related to operation during poor channel conditions. A new voice service was defined that contained multiple speech coding rates and could adapt the level of channel coding to the channel conditions. This new service became known as the Adaptive Multi-Rate (AMR) speech service for GSM.

To meet both the capacity and quality goals of the AMR service, it was defined with half and full-rate modes of operation. In the full-rate mode, there are 8 speech codec rates defined. Each includes an associated channel coding scheme. For the half-rate mode, there are 6 speech codec rates defined, each having a unique channel coding scheme. Hence, there are a total of 14 channel codes defined for AMR voice and 8 speech rates. The 6 AHS speech rates are a subset of the 8 AFS rates.

Not all of the codec modes may be used within a given call. Specifically, at call setup AMR configurations are downloaded to the MS and BTS. The AMR configuration includes an Active Codec Set (ACS) together with thresholds and hysteresis values. The ACS may contain anywhere from 1 to 4 codecs. The thresholds and hysteresis values are used by an AMR receiver to determine the optimal receive link codec mode from those within the ACS.

The advantage of AMR stems from its ability to dynamically adapt channel coding to meet the current needs of the link wherein the link may include degradations due to low signal, fading, shadowing, noise, etc. This link adaptation is assisted by measurements within the AMR receiver of both the BTS and MS. The general operation of AMR link adaptation is shown in the block diagram of **FIGURE 1**.

With respect to the MS, its receiver is required to constantly monitor channel quality in order to determine an appropriate downlink codec mode. The channel quality is quantified as a logarithmic (dB) C/I ratio. It is typically measured on a TDMA burst basis or a speech frame basis and then filtered to remove fast-varying random components. The filtered channel quality is compared against the BSS commanded threshold and hysterises to determine the optimal codec mode. The resultant mode is encoded as a Codec Mode Request (CMR) and returned to the BSS in the reverse link.

Normally, the BSS will grant the request and use the requested mode for encoding the downlink channel to the MS.

A similar procedure is followed within the BTS. Specifically, the BTS receiver monitors the uplink channel quality from the MS and determines an optimal codec mode based on threshold/hysterisis values together with potential constraints from the network control. The resultant mode is transmitted in the downlink to the MS. This mode is termed the Codec Mode Command (CMC) and is similar to the CMR with the notable exception that the CMC commands the MS as to which rate to use on the uplink whereas the CMR requests that the BTS use a rate on the downlink.

Rate adaptation must occur in a relatively fast fashion in order to be effective and, hence, is signaled using inband data encoded within each AMR traffic frame. Every frame includes inband data but it alternates in meaning between describing its host link and commanding/requesting a mode for the opposite link. When representing its host link, this data is termed the Codec Mode Indication (CMI) and it indicates how that link was encoded. A given CMI value is associated both with the frame in which it was encoded and the succeeding frame. When representing the opposite link, this data provides the CMC (transmitted in the forward link) or the CMR (transmitted in the reverse link). Regardless of its meaning, the inband data always represents two source bits (0 to 3) and can be thought of as an index into the ACS.

With respect to channel coding, the allocation of bits between speech, inband data, FEC, and CRC error protection bits is summarized in the diagram of **FIGURE 2** for both AFS and AHS frames. For each AFS frame, 8 bits are allocated for encoding the 2-bit inband data. This coding is effectively a ¼ rate block code. For AHS frames, 4 bits are allocated for the encoded inband data effecting a ½ rate block code. The speech bits are subjectively ordered and broken into three classes according to their importance. Class 1a (most important) bits have a 6-bit CRC calculated and appended to them. The class 1a bits, class 1a CRC bits, and class 1b bits are encoded using a systematic, punctured, recursive convolutional code. Any remaining speech bits are classified as class 2 and receive no channel coding. There are no class 2 bits for AFS frames as all speech bits are protected. The channel coded AMR frames are block diagonally interleaved and mapped onto bursts in the same manner as existing (HR, FR, EFR) GSM speech frames.

Given the aforementioned coding schemes, it remains to be determined how a receiver turns RF information into bits appropriate for the speech decoder and, ultimately, pleasing audio for the listener, e.g. MS user.

The GSM standard allows considerable flexibility regarding receiver design. The transmit side, particularly channel encoding scheme and related, is precisely specified while the receive side is restricted only by performance limits regarding sensitivity and the like. MS and BTS manufacturers are thus allowed alternative designs according to their appropriateness within a given architecture. For example, poor RF receiver performance may be compensated by a good baseband receiver (channel decoding) and vice versa. It is to be understood that the receiver described herein is typical and that the novel aspects of the invention are applicable to alternate receiver designs.

FIGURE 3 provides a block diagram of a typical AMR baseband receiver. RF samples, e.g. I/Q, are collected for bursts of data and passed to an equalizer/demodulation block **302**. The equalizer block typically outputs soft bits corresponding to the demodulated data. These bursts are accumulated into blocks of data corresponding to speech frames where 4 bursts comprise an AFS block and 2 bursts comprise an AHS frame. The data blocks are de-interleaved **304** and passed to the AMR channel decoder **306** for processing as well as a frame classification block **308**. Provided the data block represented speech (or comfort noise during DTX periods), the resultant speech frame output is input to the speech decoder which converts the data into PCM samples appropriate for converting into audio.

Some other data paths are also possible out of the channel decoder. Specifically, the frame classification procedure analyzes each frame out of the deinterleaver to determine its type, e.g. speech **310**, FACCH **312**, RATSCCH **314**, SID_UPDATE, etc. The resultant classification determines how the channel decoder should be run, i.e. which channel coding scheme should be decoded.

The block diagram of **FIGURE 4** describes the dataflow of the AMR channel decoder for speech frames in more detail. Received blocks of data first have the encoded inband bit field extracted. This data is block decoded **402** to determine the 2-bit source data. The decoding is accomplished by finding the codeword that is closest to the received sequence, i.e., the code word that is closest in a squared distance sense to the received sequence. This is typically done using soft received bits. The 2-bit source data indicated by the codeword is output **404** from the inband decoder.

For frames corresponding to the CMR/CMC phase, the inband bits are passed out of the channel decoder **406** for use on the opposite link. For the remaining CMI-phase frames, the inband bits are used to determine how the associated current (and next) frame should be channel decoded **408**.

The source inband data is a 2-bit index into the ACS with a maximum value of *ACS_size - 1*. The 2-bit index corresponding to the CMI is mapped to an absolute AMR mode from the entire codec set, i.e. a 3-bit value from 0 to 7 for AFS and 0 to 5 for AHS. This absolute form of CMI is used to determine which channel decoding to perform.

The next steps involve channel decoding the frame according to the absolute CMI inband data of the current or previous frame. First, the encoded data, stripped of the inband data portion, is convolutionally decoded **410** to remove channel-induced bit errors. This is typically done using a recursive Viterbi (maximum likelihood) decoder operating on soft-bit data. The resultant (hard-bit) output data includes a 6-bit CRC field. The next step involves checking the CRC against the original source data **412** to ensure the CRC is correct and removing the CRC bits from the bitstream.

The outputs of the channel decoder are a speech frame, a bad frame indication **414** derived from the CRC status (and possibly other inputs), and the codec mode that also indicates the speech rate to decode. The standard also allows for the classification of a frame as a "degraded frame" if the CRC passes but other parameters indicate that the frame is unreliable.

This method is also described by the flow chart of **FIGURE 5** that applies only to CMI-phase frames. Received blocks of data first have the encoded inband bit field extracted and block decoded **502** to determine the 2-bit source data. The source inband data is a 2-bit index into the ACS with a

maximum value of *ACS_size - 1*. The 2-bit index corresponding to the CMI is mapped **504** to an absolute AMR mode from the entire codec set, i.e. a 3-bit value from 0 to 7 for AFS and 0 to 5 for AHS. Next, the encoded data, stripped of the inband data portion, is convolutionally decoded **506** to remove channel-induced bit errors. The next step involves checking the CRC against the original source data **508** to ensure the CRC is correct and removing the CRC bits from the bitstream. This is followed by a bad frame metric calculation **510**. Next, a check is made to determine if the frame is good **512**. If it is, the speech frame is passed to the speech decoder **514**. Otherwise, the speech frame undergoes a second check to determine how bad the speech frame is **516**. If it is a degraded frame it is marked as such and the decoded bits are passed to the speech decoder **518**. If it is more than degraded, then the speech frame is masked with respect to the speech decoder **520**.

An important factor in user-perceived audio quality during marginal channel conditions is receiver (RF and baseband) sensitivity. This is quantified using a variety of measures. The measures of interest in this context are the Frame Erasure Rate (FER) and the Residual Bit Error Rate (RBER).

The FER refers to the rate at which frames are "erased" due to CRC failures or excessive bit errors. Such frames are not recoverable and typically require bad frame masking within the speech decoder, e.g. repetition of a previous frame or muting/comfort noise generation. RBER refers to the bit error rate which is present in the received bitstream when those frames which are erased are excluded from the statistics.

A good-performing inband decoder is necessary to achieve both low FER and RBER. For purposes of explanation, consider a marginal channel in which the inband data is decoded incorrectly. For any such frame, the wrong channel decoder will be run leading to a frame erasure or (on rare occasions) very high RBER.

Inband bit decoding problems are more pronounced due to the fact that the channel codes are relatively strong. For example, in AFS service the lowest speech codec mode (4.75 kbps) is coded using a 1/5 rate recursive systematic convolutional code which is punctured to an effective rate of 101/442. The corresponding inband data is coded using a simple ¼ rate block code. Likewise, the lowest AHS speech service is coded using a recursive systematic punctured rate 1/3 convolutional code whereas the inband data is coded using a simple rate ½ block code. For these and other low rates, the channel codes for the payload data have more error-correcting capability than those of the inband data bits. In other words, in marginal channel conditions the channel decoder (Viterbi and CRC check) may be capable of salvaging many of the frames (correcting the errors/minimizing the BER & FER) provided the inband decoding commands that the correct channel decoder run. However, the inband decoder will tend to fail often and the normal channel decoder will not get the chance to salvage bad frames.

There are a couple of issues that further compound the aforementioned inband decode problem. First, the relatively weak inband coding and strong channel coding occur at the lower operational modes, e.g. 4.75 kbps speech. It is at these rates that the problems are most likely to occur. Such lower rates are used when the channel is quite poor and the combination inband/channel decode

needs to perform its best. Second, due to the fact that a given frame's CMI controls the current and next frame, a bad decode will erase 2 frames rather than just one.

Performance degradation due to bad inband decodes can be reduced by using a priori knowledge when performing the decode, e.g. using Markov modeling with statistical information. In qualitative terms, the CMI inband data does not often change. It is derived from a channel quality measure which is typically heavily filtered and associated with threshold values. Provided adequate hysterisis values are used, the filtering effectively prevents many mode changes from occurring, e.g. mode changes would typically occur with a mean-time between changes on the order of seconds. Hence, for a given CMI-phase frame, the inband data is most likely to stay the same as that previously decoded. By biasing the inband decoder to stay in the same state, its performance can be significantly increased.

Disclosure of the Invention

The present invention comprises a system and method for channel decoding speech frames in a receiver capable of multiple (M) codec modes, wherein channel encoded speech frames are comprised of at least an inband bit portion and a speech portion. An inband bit decoder decodes the inband bit portion of a received frame to obtain confidence levels associated with each of the M codec modes. Using these confidence levels, the codec modes are ordered from most to least likely. The speech frame is then decoded by a channel decoder using the most likely codec mode. A frame determination check is performed to determine the quality of the decoded speech frame. If the decoded speech frame is determined to be of poor quality, then the channel decoding process is repeated using the next most likely codec mode corresponding to the next highest inband bit decoding confidence level.

In an alternative embodiment, the channel decoding process is repeated for a maximum number of iterations wherein the upper limit of this maximum is equal to the number of codec modes (M). Moreover, the maximum number of iterations can be limited by the number of confidence levels that exceed a threshold value.

In yet another embodiment, there is disclosed a system and method of channel decoding speech frames in a receiver capable of multiple (M) speech codec modes wherein the channel encoded speech frames are comprised of an inband bit portion and a speech portion. An inband decoder calculates an inband decode metric for each codec mode. A channel decoder then partially decodes speech data for each of the M codec modes. The most likely codec mode is then determined based upon the partially decoded speech data and the calculated inband decode metric data. At this point, the decoding of the speech data is resumed using the most likely speech codec mode just determined.

Brief Description Of The Drawings

FIGURE 1 is a block diagram illustrating the high level operation of an AMR system.

FIGURE 2 illustrates AMR bit allocations.

FIGURE 3 is a block diagram illustrating a baseband AMR receiver.

FIGURE 4 is a prior art block diagram illustrating an AMR speech frame channel decoder.

FIGURE 5 is a prior art flow chart illustrating a process for channel decoding an AMR speech frame.

FIGURE 6 is a block diagram illustrating an AMR speech frame channel decoder according to the present invention.

FIGURE 7 is a flow chart illustrating a process for channel decoding an AMR speech frame according to the present invention.

FIGURE 8 is a flow chart illustrating an alternative process for channel decoding an AMR speech frame according to the present invention.

FIGURE 9 is a flow chart illustrating yet another alternative process for channel decoding an AMR speech frame according to the present invention.

FIGURE 10 is a flow chart illustrating still another alternative process for channel decoding an AMR speech frame according to the present invention.

Best Mode for Carrying Out the Invention

For simplicity and clarity, the description herein is written in the context of a GSM AMR system. However, the methods disclosed are also applicable to other multi-rate audio services, e.g. wideband AMR or the EVRC of narrowband CDMA (IS-95).

The present invention improves the performance of an AMR channel decoder primarily by means of increasing the probability of finding the correct inband data for the CMI-phase frames. The increased performance pertains to call configurations in which the AMR ACS contains multiple codecs.

The methods of the present invention may be concretely implemented in software (DSP or general-purpose microprocessor), digital hardware (e.g. ASIC or FPGA) or a combination thereof. The primary requirement for any such application is that multiple channel coding methods be defined and that there exist some method of distinguishing good and bad frame decodings.

Markov-model based inband decoding significantly improves performance at a cost of increased computational complexity. Such added complexity may not be feasible in some architectures so an entirely other method may be needed. Other architectures may not be capable of implementing an ideal Markov-based decoder and instead use a reduced complexity version that performs between that of a simple no-memory decoder and an ideal Markov-based decoder.

The method of the present invention could be used in place of a Markov-based inband decoder (i.e. with a simple no-memory decoder), in conjunction with a Markov-based inband decoder, or in conjunction with some other variant decoder. In fact, the specifics of the inband decoder are somewhat immaterial to the present invention. Most any inband decoder could be used and the present invention would improve their performance to varying degrees. The novel aspect of the invention is in higher-level control of the inband decoder and its combination with other receiver blocks.

In the receive method of **FIGURE 4**, the inband decoder, convolutional decoder, and CRC check are run sequentially. If the CRC check fails or the frame is classified as "bad" via other means, the speech decoder is informed that the frame is bad and masks it in some way. The receiver effectively gives up on the frame and the FER is adversely affected as is the speech quality.

However, if the frame failed due to bad inband decoding, the frame may still be salvageable. What is needed are additional attempts at channel decoding using the other active channel codes.

The method of the present invention is described in the block diagram of **FIGURE 6**. The difference between this diagram and that of the prior art (**FIGURE 4**) is that **FIGURE 6** includes a feedback path **602** from the bad frame determination block to the initial blocks of the channel decoding. Thus, a bad (or potentially bad) frame forces the channel decoder to run again utilizing a different channel decoding scheme.

The method of the present invention is more explicitly explained in the flow chart of **FIGURE 7**. First, the inband data is decoded **700** and the indices are ordered from most likely to least likely, e.g. based upon a Euclidean distance measure. For example, the inband decoder might have 60% confidence that a 3 was transmitted, 25% confidence that a 1 was transmitted, 10% confidence that a 2 was transmitted and 5% confidence that a 0 was transmitted leading to an order of {3,1,2,0}.

The most likely inband decoding (3 in the example) is chosen **704** and mapped **708** to its 3-bit absolute codec mode using the active codec set. For an example ACS of {4.75 kbps (absolute codec 0), 5.90 kbps (absolute codec 2), 7.95 kbps (absolute codec 4), 10.2 kbps (absolute codec 6)}, the mapping would be to the 10.2 kbps mode.

Using the chosen mode and its associated channel coding, a recursive convolutional decode is performed **712** on the class 1 portion of the non-inband received bits (typically soft). The resultant output data contains both a speech frame (204 bits for the example above) together with a 6-bit CRC. The CRC field is extracted and checked **716** against the correct value for the class 1a bits which have been output by the convolutional decoder.

Next, a bad frame classification procedure **720** is performed. For the simplest case, this would consist of only a CRC check, i.e. if the CRC check fails, the frame is classified as bad whereas if it passes the frame is classified as good. Due to the weakness in the CRC and the necessity to classify frames as "degraded," such a simple check is generally not adequate so another check is needed even when the CRC passes.

One such method is to re-encode the channel decoded data and compare it against the originally received data to estimate a channel BER. This method is commonly used in GSM (with the fixed-rate speech services) as the mechanism is usually already in place in order to calculate RXQUAL. If the resultant BER estimate is high, the frame is classified as "bad," if it is low the frame is classified as "good," and if it is moderate the frame is classified as "degraded." The precise method for bad frame determination is considered outside of the scope of this invention and it should be recognized that most current (or future) methods for such classification could be used within this invention.

If the result of the bad frame determination block is that the frame is good **724**, those bits corresponding to the speech frame (204 for the 10.2 kbps mode) are passed to the speech decoder **728** along with a flag that the associated frame is good. The channel decode procedure is then complete for that frame.

If, however, the frame is not classified as good a different procedural path is taken. First, it is checked that the number of iterations **732** through the loop has not reached the iteration threshold *N*. For optimal performance, this threshold is set to the size of the ACS, e.g. 4 in the example previously described. In practice, the performance gain for each additional iteration decreases significantly. Each iteration involves a convolutional decoding which is not computationally trivial meaning many implementations (particularly software-based) will not be capable of such an exhaustive search. For such implementations, *N* is set lower than the ACS size to a value such as 2.

If another iteration is to be taken, the next most likely inband decoding is chosen **736** to be mapped to an absolute (3-bit) codec mode. The procedure then continues as described previously starting with the inband data mapping. For the example described above, the inband decoding chosen would be 1 which would then be mapped to absolute codec 2 (5.90 kbps mode).

If the maximum number of iterations has already been reached, then the channel decode loop exits. A check is made to see if the last channel decoded frame is bad **740** or not. This generally involves checking a metric calculated as part of the bad frame determination block previously. If the frame is bad and there were no previously channel decoded frames flagged as degraded **748**, the speech decoder is informed that it needs to mask the frame **752**. If the frame is not bad (and not good per the previous check), it is flagged as degraded **744** and sent to the speech decoder.

It may happen that in attempting to find a "good" frame in the multipass channel decoding a "degraded" one was overlooked. Hence, it is necessary to maintain the grading of each channel decoding attempt. If the maximum iteration count is reached with no "good" frame found, a search is made to see if any of the decodings yielded a degraded frame. If so, that frame is flagged appropriately **756** and passed on to the speech decoder.

If only the most likely inband decoding has a significant metric, it is useful to exit the loop early. Consider an example with 3 active codecs and with inband decoding likelihoods {96%, 2%, 2%}. If the bad frame determination classifies the frame as not good on the first iteration, it is quite unlikely that another pass through the loop will find a good frame. In this case, the inband data is probably good but the rest of the payload has significant errors causing a bad CRC or other bad frame indication.

This submethod serves to reduce the computational complexity of the preferred method but does not significantly affect performance. The flow chart of **FIGURE 8** provides an example. It is implemented by adding a threshold metric check **802** somewhere between the "Good Frame" check and the entry back into the channel decode loop at the "Map Inband Data" block. The new switch **802** checks if the inband data to drive the next channel decode was decoded with a reasonable confidence, i.e. above some threshold. If not, the loop exits and execution enters the "Bad Frame" check **740** after

the iteration check. Note that the specific ordering of blocks after the "Good Frame" check in the flow chart may be changed to achieve equivalent results. For example, the iteration check could be moved after the metric check.

It should be recognized by one skilled in the art that an equivalent method may also be implemented early in the process flow by checking the inband decode confidence levels after their ordering and setting the iteration threshold accordingly. Such a process is shown in the flow chart of **FIGURE 9**. For example, the algorithm may determine that only inband data having a confidence level greater than 20% should be considered and the iteration threshold *N* is set to ensure this **902**. For the previous example with confidences {60%, 25%, 10%, 5%}, this means *N* would be set to 2 and only modes corresponding to inband data values of 3 and 1 will be considered for channel decoding. This process also shows that it may be necessary to limit the iterations **904** determined by the procedure. Though this method is generally equivalent performance-wise to the previous one, it has certain implementation advantages that may make it preferable in some architectures.

Henceforth, an alternate embodiment is presented that provides similar performance to the embodiment described with respect to **FIGURE 7**. The alternate embodiment moves the check for unreliable frames to early in the process so that multiple complete channel decodes may be avoided. This method is not as elegant as that of the preferred embodiment but this alternative may be more appropriate for implementations requiring lower worst-case computation.

As stated in the background, the channel decoder is typically implemented using a Maximum Likelihood Sequence Estimation (MLSE) technique commonly referred to as a Viterbi decoder. The reader is presumed to be at least moderately familiar with Viterbi decoding techniques including their usage with recursive channel codes. Though this description is written in the context of AMR which uses such recursive codes, the unique aspects of the invention apply equally to other systems which use non-recursive convolutional codes. Other decoding techniques are possible but this description presumes usage of the Viterbi algorithm.

The Viterbi algorithm is usually described using a trellis. Each column in the trellis is referred to as a *stage* whereas each node within a given column is referred to as a *state*. The first stage of the trellis has a single state. The number of states in each subsequent stage doubles up until the size reaches 2 to the power of the (constraint length –1) where the constraint length is defined as the number of positions in the channel encoder shift register. The number of states stays constant until near the end of the trellis. Typically, the shift register in the convolutional encoder is flushed with zeroes once all of the valid data bits have been encoded. This is represented in the trellis by its contracting from its steady-state size back down to a single state where the number of states is halved at each stage, i.e. the end of the trellis looks like the mirror image of the start of the trellis. For a rate 1/n code, a given state in the trellis initially leads to 2 states in the next stage meaning each state in the next stage has two input paths. For each state in the next stage, only the transition leading to the best metric at that state is retained, i.e. one of the 2 transitions gets pruned.

In tracing through the trellis using the Viterbi algorithm, a metric is calculated for each state within a given stage. This metric, henceforth referred to as a Viterbi metric, indicates the confidence level that the transmitted data corresponds to that of the trellis path leading to that state. In practice, the metric need not be maintained for every node in the trellis; it is only necessary to maintain a metric for the maximum number of states at a given stage. Such metrics may be normalized at each stage when transitioning through the trellis but for purposes of this embodiment the metrics should be accumulated without normalization. The best metric at a given stage represents the most likely path up to that point.

The alternate embodiment is described in the flowchart of **FIGURE 10**. In this method, the inband data is analyzed and a metric is calculated **1002** for each possible mode, e.g. the Euclidean distance is measured to determine a confidence level for each mode. A Viterbi decode is performed **1004** for each possible channel coding (mode) only up through a certain stage. The number of stages traversed should be large enough to achieve some confidence that the Viterbi metric at that stage is representative but not so large that the computational advantage of the method is lost. The number of stages should be at least equal to the number stages required to reach the steady-state for each code, i.e. the *constraint length – 1* of the code with the largest constraint length. A value of 2-3 times said constraint length is recommended, e.g. 20 stages might be traversed.

At this point, the best metric is found for each channel decode attempt **1006**. This is mapped to a confidence level which is combined with the inband decode results to determine the most likely channel mode **1008**. The Viterbi decode corresponding to that mode is then restarted from the point at which it prematurely stopped. The resulting decoded frame is classified and speech decoded in the normal manner.

A slight variation on this method involves ordering the inband decode metrics and performing the partial Viterbi decodes in the order indicated as most likely by the inband decoder. If the best Viterbi metric for the most likely mode is above some threshold, the channel decode continues and that mode is definitively used. Otherwise, a partial Viterbi decode is performed assuming the next most likely mode. If the best Viterbi metric for that decode is above some threshold, the channel decode continues for that mode. This process repeats until a best partial Viterbi metric is found which is above the threshold or the possible modes have been exhausted. If the decodings are exhausted, the one with the best metric is pursued.

It is to be recognized by one skilled in the art that various combinations of the methods described above may also be used along with derivatives that are not explicitly discussed. The methods could be implemented in software (DSP or general-purpose microprocessor), hardware, or a combination.

It should also be noted that the term "receiver" as used herein refers to the receiving portion of a cellular transceiving device. A cellular transceiving device includes both a mobile terminal (MS) as well as a base station (BSS). A mobile terminal must be in communication with a base station in order to place or receive a call. There are numerous protocols, standards, and speech codecs that can be used for wireless communication between a mobile terminal and a base station.

While the present invention is described herein in the context of a mobile station, the term "mobile station" may include a cellular radiotelephone with or without a multi-line display; a Personal Communications System (PCS) terminal that may combine a cellular telephone with data processing, facsimile and data communications capabilities; a Personal Digital Assistant (PDA) that can include a radiotelephone, pager, Internet/intranet access, Web browser, organizer, calendar and/or a global positioning system (GPS) receiver; and a conventional laptop and/or palmtop receiver or other computer system that includes a display for GUI. Mobile stations may also be referred to as "pervasive computing" devices.

Specific embodiments of the present invention are disclosed herein. One of ordinary skill in the art will readily recognize that the invention may have other applications in other environments. In fact, many embodiments and implementations are possible. The following claims are in no way intended to limit the scope of the present invention to the specific embodiments described above. In addition, any recitation of "means for" is intended to evoke a means-plus-function reading of an element and a claim, whereas, any elements that do not specifically use the recitation "means for" are not intended to be read as means-plus-function elements, even if the claim otherwise includes the word "means".